



# ***Sound Classification and Localization***

***Based on***

## ***Biology Hearing Models and***

## ***Multiscale Vector Quantization***

**John S. Baras**

***Center for Auditory and Acoustic Research  
Electrical and Computer Engineering Department  
and the Institute for Systems Research  
University of Maryland College Park***

<b>Report Documentation Page</b>			Form Approved OMB No. 0704-0188	
<p>Public reporting burden for the collection of information is estimated to average 1 hour per response, including the time for reviewing instructions, searching existing data sources, gathering and maintaining the data needed, and completing and reviewing the collection of information. Send comments regarding this burden estimate or any other aspect of this collection of information, including suggestions for reducing this burden, to Washington Headquarters Services, Directorate for Information Operations and Reports, 1215 Jefferson Davis Highway, Suite 1204, Arlington VA 22202-4302. Respondents should be aware that notwithstanding any other provision of law, no person shall be subject to a penalty for failing to comply with a collection of information if it does not display a currently valid OMB control number.</p>				
1. REPORT DATE <b>24 AUG 1999</b>	2. REPORT TYPE <b>N/A</b>	3. DATES COVERED <b>-</b>		
<b>4. TITLE AND SUBTITLE</b> <b>Sound Classification and Localization Based on Biology Hearing Models and Multiscale Vector Quantization</b>			5a. CONTRACT NUMBER	
			5b. GRANT NUMBER	
			5c. PROGRAM ELEMENT NUMBER	
<b>6. AUTHOR(S)</b>			5d. PROJECT NUMBER	
			5e. TASK NUMBER	
			5f. WORK UNIT NUMBER	
<b>7. PERFORMING ORGANIZATION NAME(S) AND ADDRESS(ES)</b> <b>University of Maryland</b>			8. PERFORMING ORGANIZATION REPORT NUMBER	
<b>9. SPONSORING/MONITORING AGENCY NAME(S) AND ADDRESS(ES)</b>			10. SPONSOR/MONITOR'S ACRONYM(S)	
			11. SPONSOR/MONITOR'S REPORT NUMBER(S)	
<b>12. DISTRIBUTION/AVAILABILITY STATEMENT</b> <b>Approved for public release, distribution unlimited</b>				
<b>13. SUPPLEMENTARY NOTES</b> <b>DARPA, Air-Coupled Acoustic Microsensors Workshop held on August 24 and 25, 1999 in Crystal City, VA., The original document contains color images.</b>				
<b>14. ABSTRACT</b>				
<b>15. SUBJECT TERMS</b>				
<b>16. SECURITY CLASSIFICATION OF:</b>			<b>17. LIMITATION OF ABSTRACT</b> <b>UU</b>	<b>18. NUMBER OF PAGES</b> <b>15</b>
a. REPORT <b>unclassified</b>	b. ABSTRACT <b>unclassified</b>	c. THIS PAGE <b>unclassified</b>		



# Motivation



- **Algorithms motivated by similar processing in animals and humans:**
  - Hearing and sound classification
  - Vision and identification of objects
- **Text-independent robust speaker identification**
  - Identifying the speaker from the “music” of his voice
- **Speaker-independent speech recognition**
  - Identifying phonemes, vowels, words from their inherent sounds
- **Identification of musical instruments (“timbre”)**

## Applications to acoustic signal recognition

- Fault identification in tools and wear prediction
- Ground vehicle identification from array microphones

**NEXT CHALLENGE: Biology Inspired Sensor Network processing**



# **Acoustic Vehicle Classification**

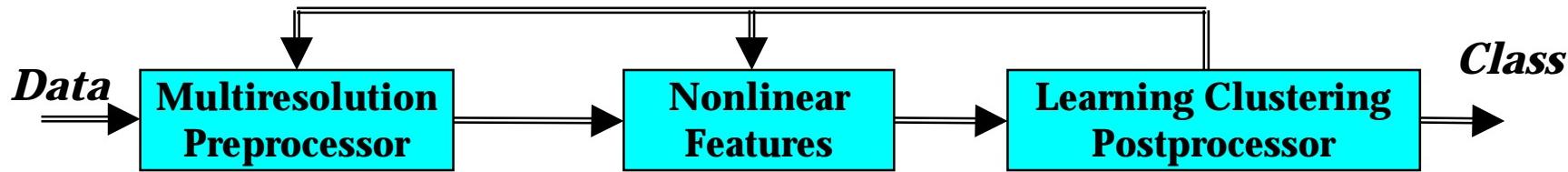
## **Objectives and Challenges**



- Develop systematic methodologies and algorithms; not *ad hoc*
- Robust Target ID (wrt environment, terrain, speed)
- Algorithms for combined DOA (localization) and target ID
  - Localization assisted ID
  - ID assisted localization
- Multi-target detection, ID and DOA; separation of closely spaced targets
- Robust feature extraction from auditory models; dynamic DOA and ID
- Algorithm evaluation in the field and comparison against conventional algorithms for detection, DOA and ID

- **Architecture**

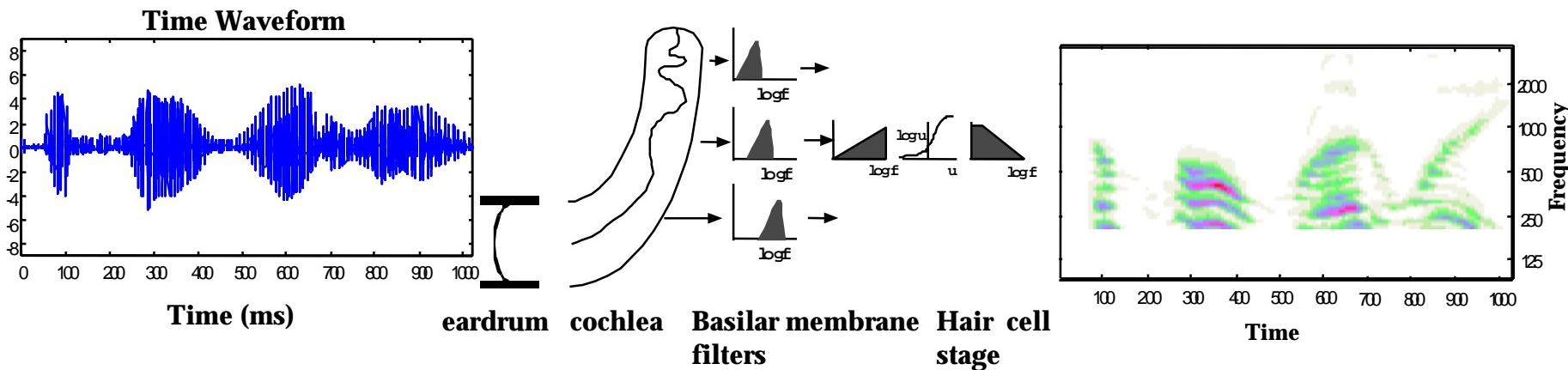
*Feedback*



- Architecture and formulation address two most important issues:
  - Progressive classification; Which features to use and when
  - Efficient design of databases for reference signals and fast search
- Trade-off between efficiency in features (compression) and accuracy in classification leads to
- Mathematical formulation of the problem:
  - Combined compression and classification for general signals
  - Content-based feature extraction and use for classification

# Multiresolution Preprocessor: Auditory Filtering

Two auditory filters, motivated and designed according to acoustic physiology and acoustic cortex models, were used to compute the timbre spectrogram of one particular subframe in each frame



- The first filter mimics the action of the inner ear
- Computes the spectrogram of the sound sample, and performs various nonlinear operations, which models the nonlinear fluid-cilia couplings and ionic channels of conduction

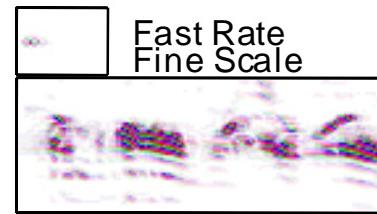
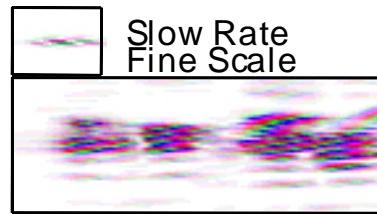
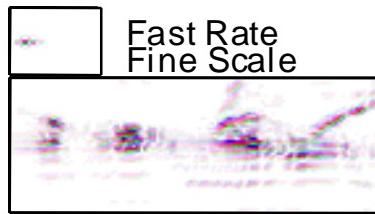
( Wavelet Transform )



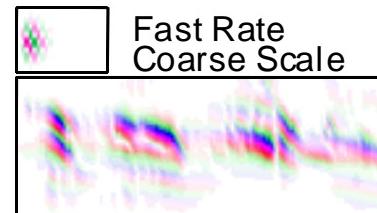
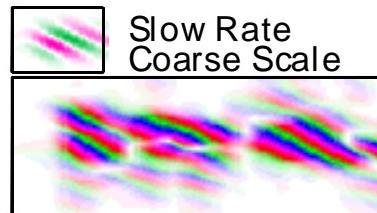
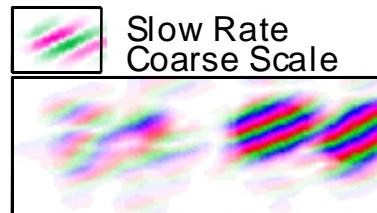
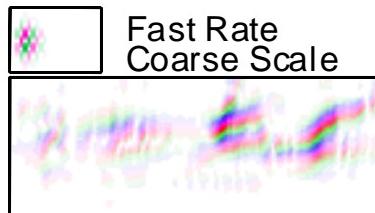
# Multiresolution Preprocessor: Auditory Filtering



## Multiresolution cortical filter outputs

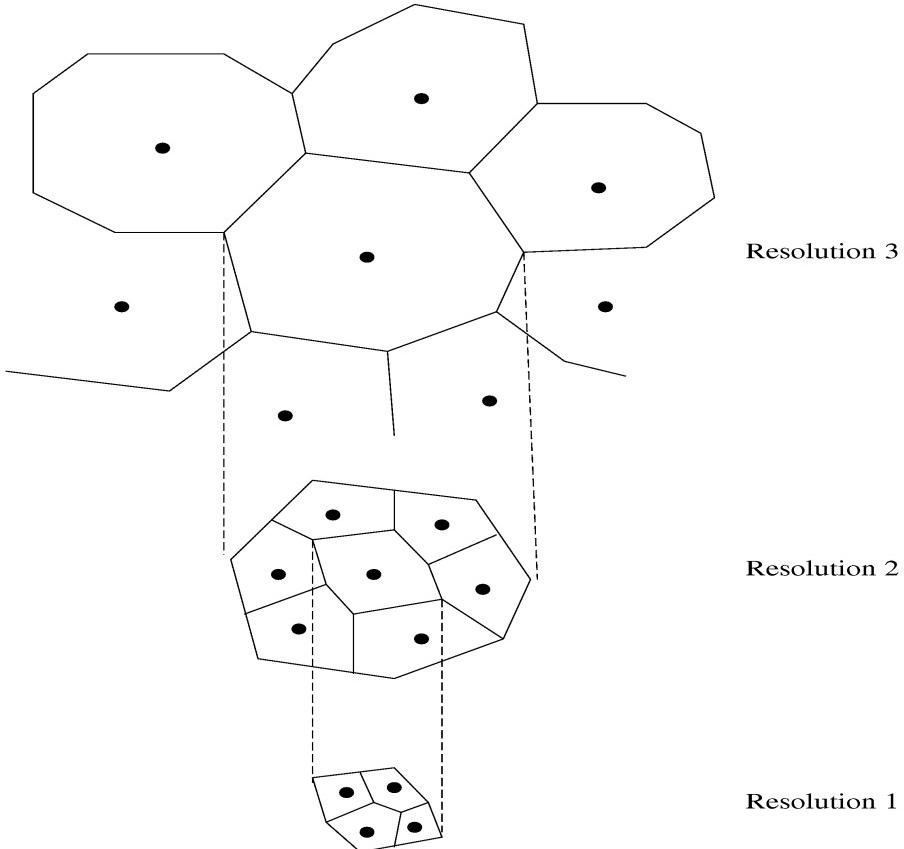


Upward Moving



Downward Moving

- The second filter models the multiscale processing of the signal that happens in the auditory cortex
- A Ripple Analysis Model, using a ripple filter bank, acts on the output of the inner ear to give multiscale spectra of the sound timbre (**Wavelet Transform**)

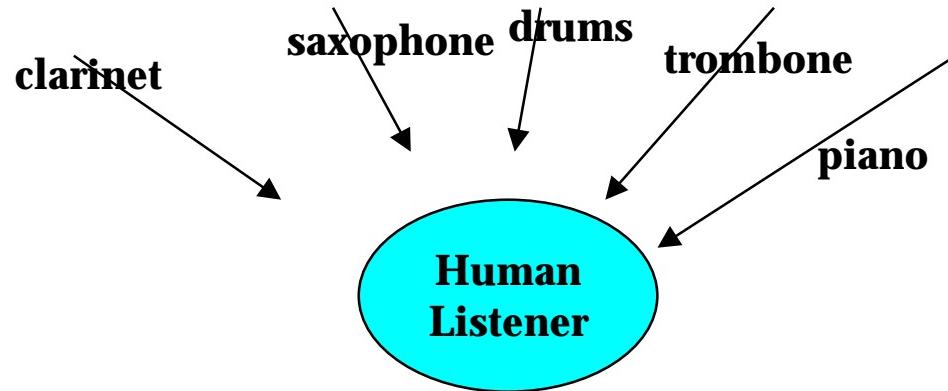


- First perform a multiresolution wavelet representation of the signals
- Consider each signal  $f$  at different resolutions  $S^0 f, S^1 f, \dots, S^{J^*} f$
- Proceed by partitioning the signal space at various resolutions in progressively finer cells
- **Greedy algorithm** works by splitting the cell with maximum distortion using finer resolution data

Layer in tree  $l = J^* - m$ ,  $m$  the scale (top layer 0: coarsest)  
 Cell labels: (layer, index) or (scale, index)

# Approach

**Can we mimic and understand the ability of humans to do partial recognition of musical instruments and DOA in a combined and mutually enhancing fashion?**



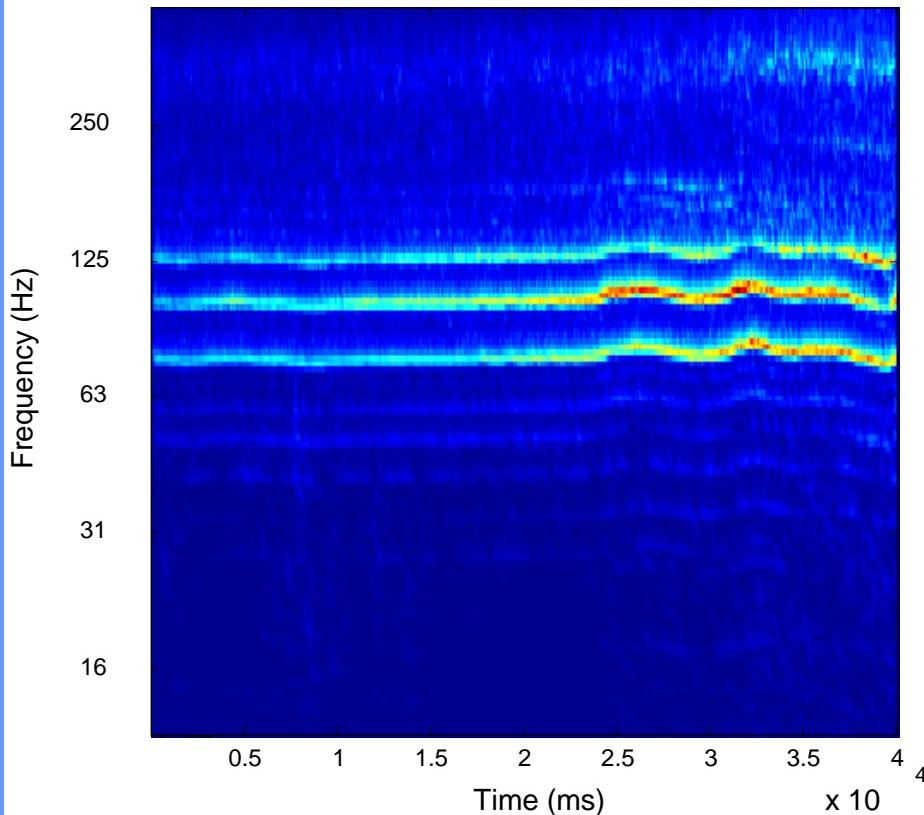
- Combine the Stereausis model and its derivatives , with the Auditory filtering multiscale VQ algorithms
- Using the cochlea, cortical, or combined spectra, perform DOA on a “per frequency band basis”
- Combine portions of spectra according to DOA
- Use the multiscale classifier to ID portions of spectra tagged by angle, as compared to stored vehicle spectra
- Repeat the cycle as the scenario evolves



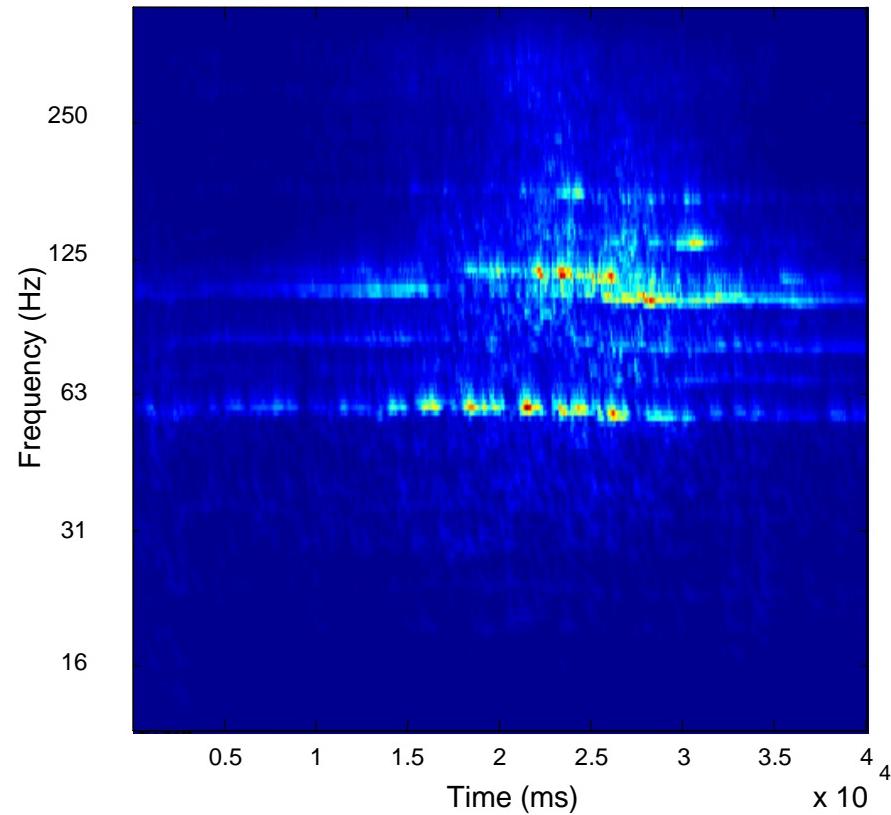
# Auditory Processing of Vehicle Acoustic Signals: Cochlea



gv1a1012.mat: type 1 speed 5 desert



gv1b2021.mat: type 1 speed 10 arctic

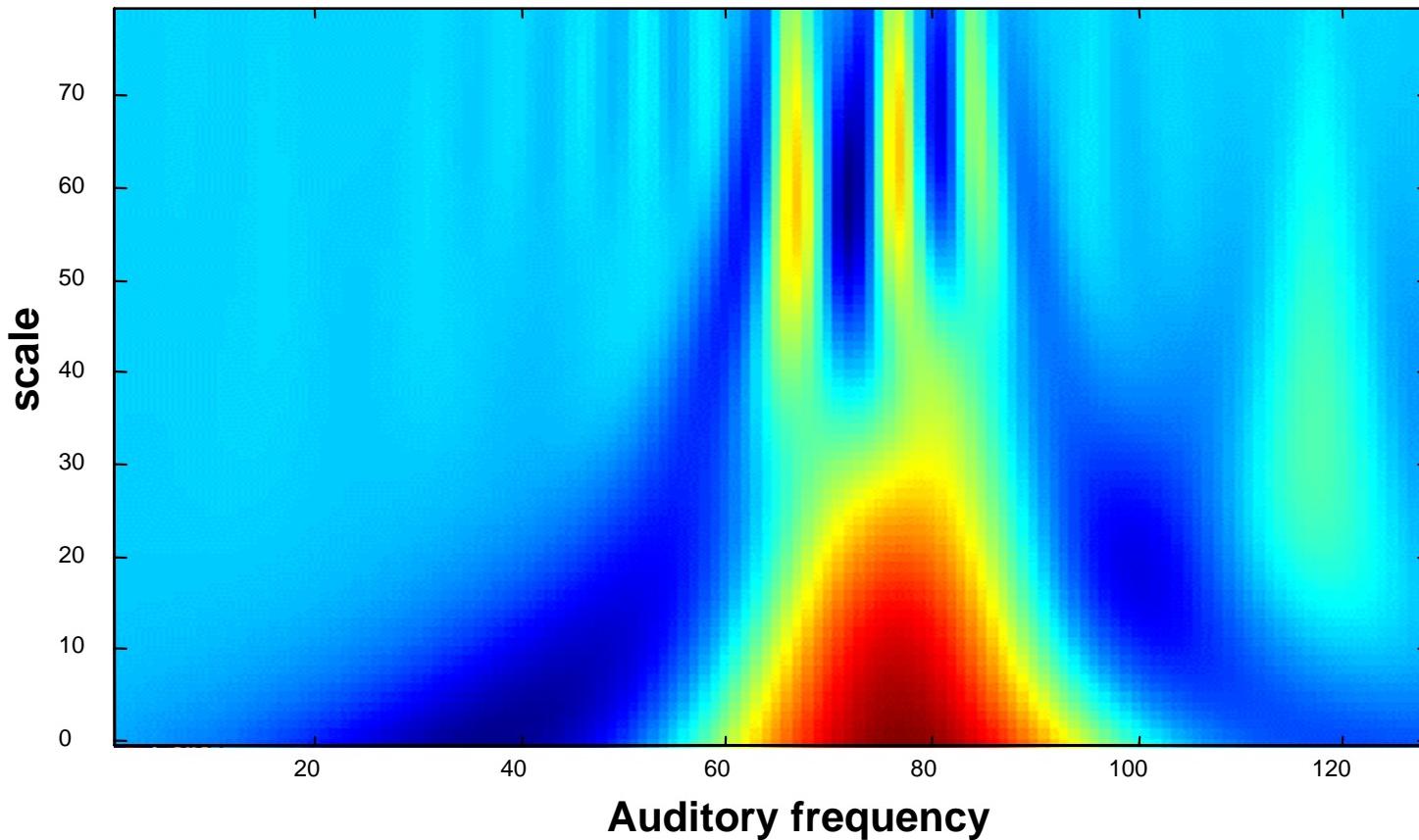


**Auditory processing for vehicle signals (cochlear filter banks)**

**Left: vehicle type 1, speed 5km/hr. Right: vehicle type 1, speed 10km/hr**

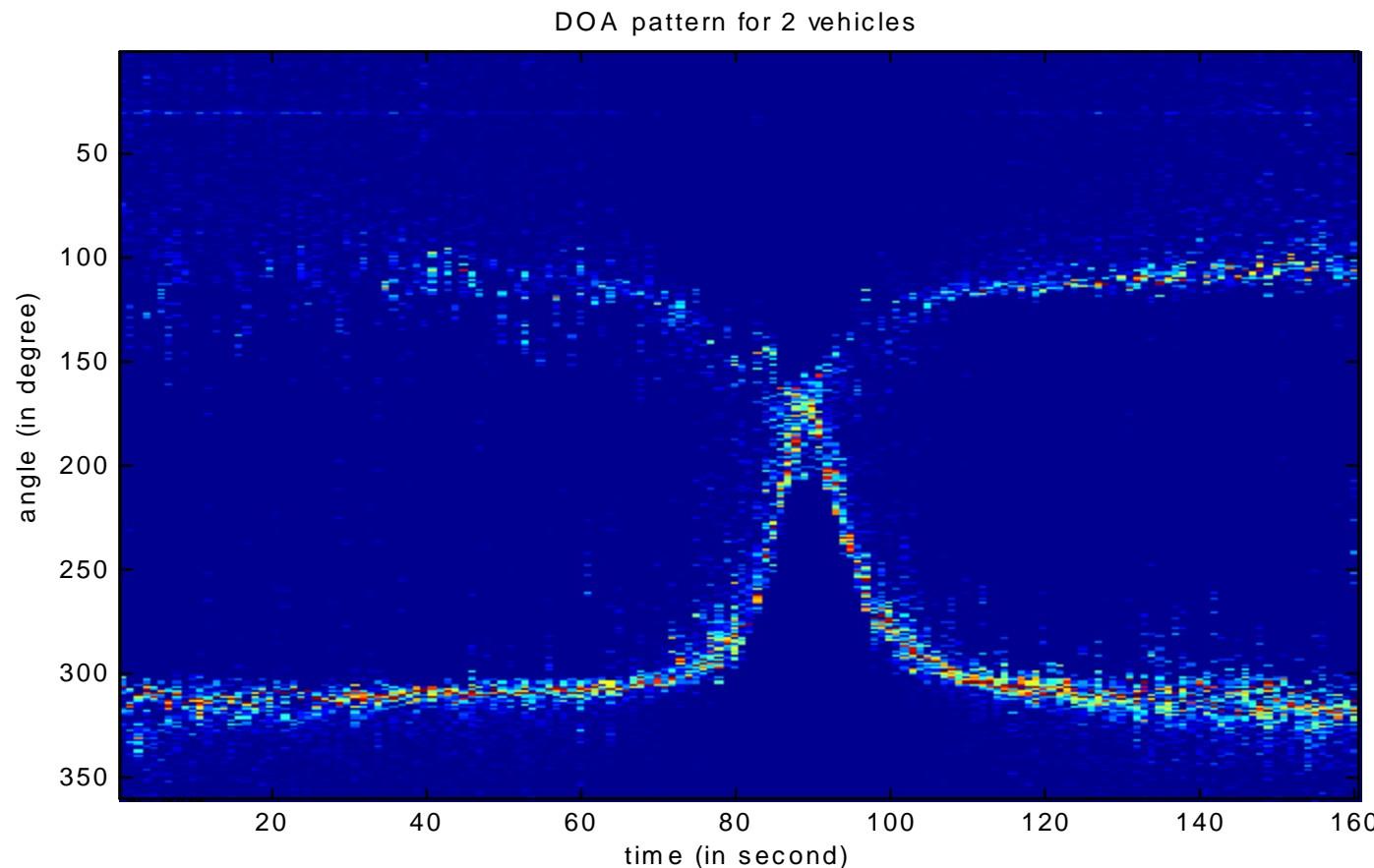


# Auditory Processing of Vehicle Acoustic Signals: Cortex



Example of multi-resolution representation from  
cortical module

# Stereausis Output for Two Vehicles



Relatively easy case: Large angular separation between two vehicles

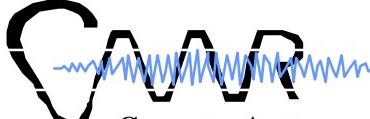
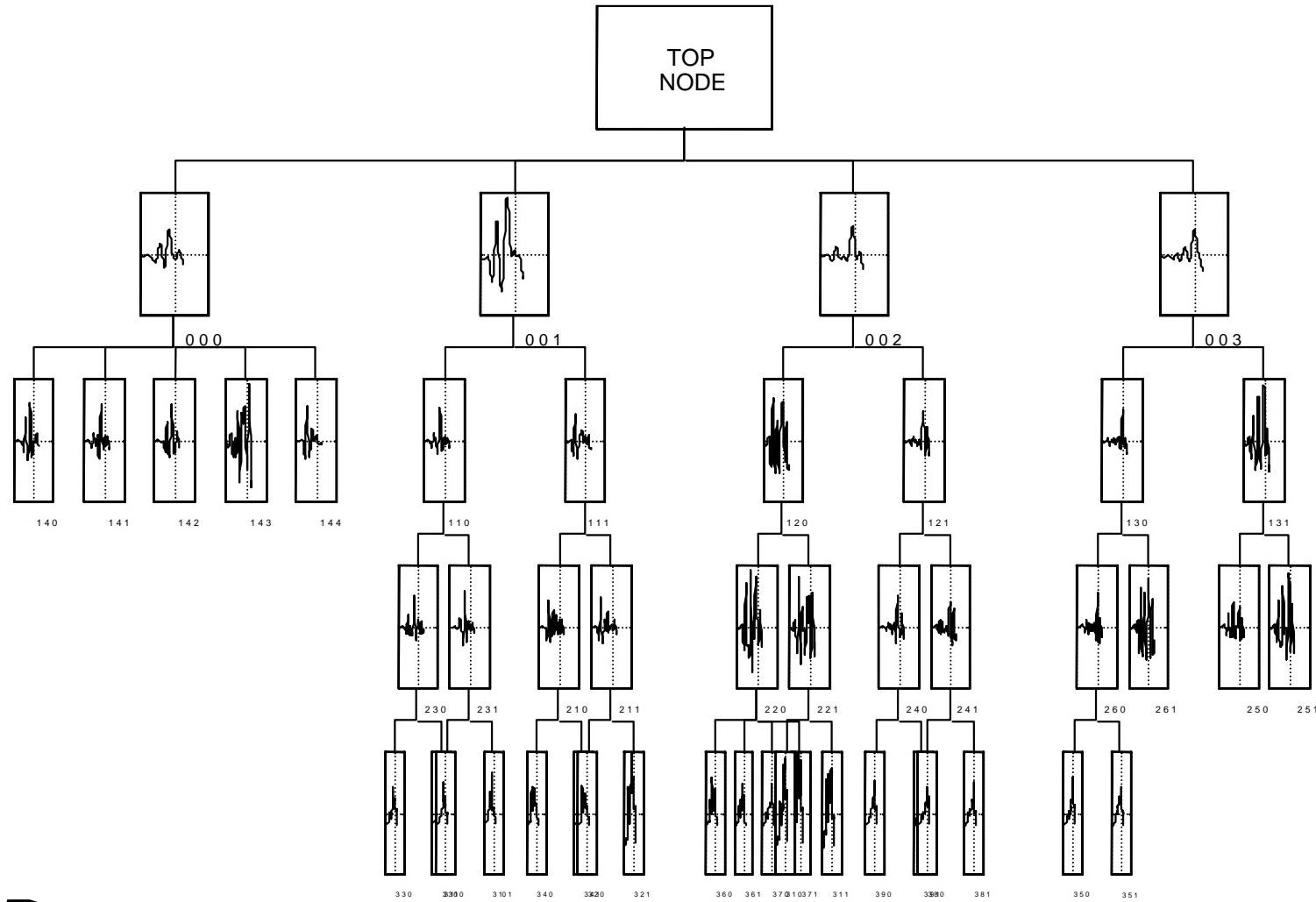


## *Leaf Node Entropies for PTSVQ Tree of Vehicle Type 8*



# cell entropy

1	4	0	1.3570
1	4	1	0.9503
1	4	2	1.1779
1	4	3	1.0735
1	4	4	1.3022
2	5	0	0.6365
2	6	1	0
3	1	0	0.5765
3	1	1	0.2993
3	2	0	0.7516
3	2	1	0.4765
3	3	0	0.7633
3	3	1	0.5670
3	4	0	0.4540
3	4	1	0.4384
3	5	0	0.2728
3	5	1	0.4975
3	6	0	0.5313
3	6	1	0.3061
3	7	0	0.6054
3	7	1	0.6383
3	8	0	0.4824
3	8	1	0.5377
3	9	0	0.5044
3	9	1	1.2556
3	10	0	1.0144
3	10	1	1.1967





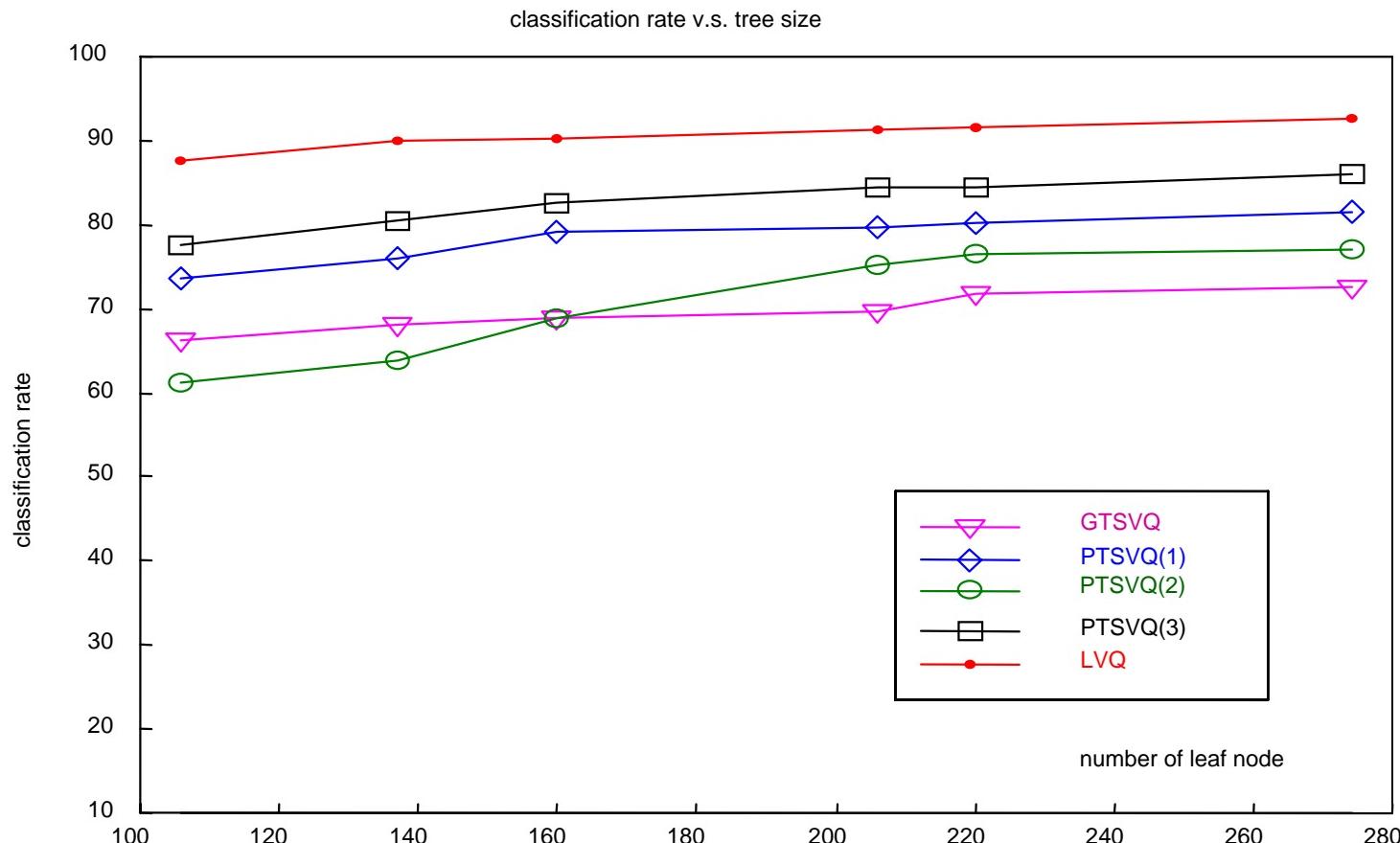
# Options in Applying WTSVQ to Acoustic Vehicle Classification



- **GTSVQ:** A global tree-structured multi-resolution clustering mechanism that mimics the aggressive and topological hearing capabilities of biological systems. Here a global tree is built on training data from all vehicles. **New vehicle insertion problem.**
- **LVQ:** A supervised learning neural network, LVQ achieves optimal classification in the Bayes sense. It has the disadvantages of a long search time and sensitivity to initial conditions.
- **Parallel TSVQ (PTSVQ):** build one (or more) trees for each vehicle. It achieves a trade-off between GTSVQ and LVQ on classification performance and search time. **Easy new vehicle insertion.**
- The following node allocation schemes are examined for PTSVQ:
  - PTSVQ(1): Allocation based on sample a priori probability
  - PTSVQ(2): Allocation based on equal distortion
  - PTSVQ(3): Allocation according to vehicle speed

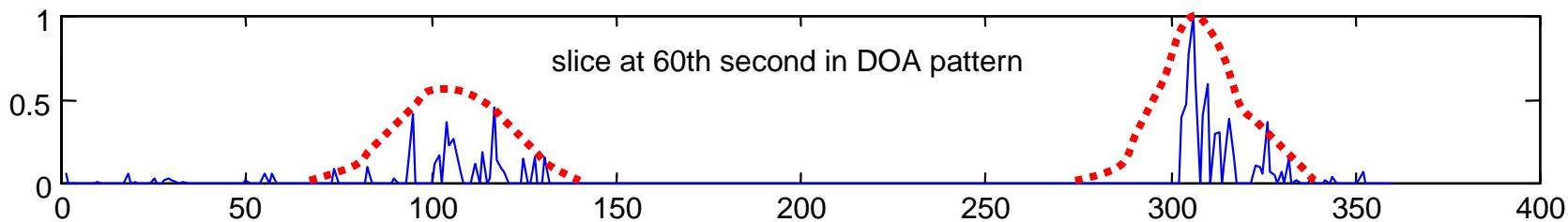


# Performance Comparisons among Options



Classification Performance: 70% samples for training, 30% for testing (same microphone)

- Angular position of each peak corresponds to DOA estimate from each cochlea band
- Can use up to 128 bands
- Amplitude indicates signal energy in the band



- Low pass filtering is performed on groups of band amplitudes and the resulting peak is used as the DOA estimate for the vehicle
- Cluster according to angular position of peaks: spectral portions tagged by angle